

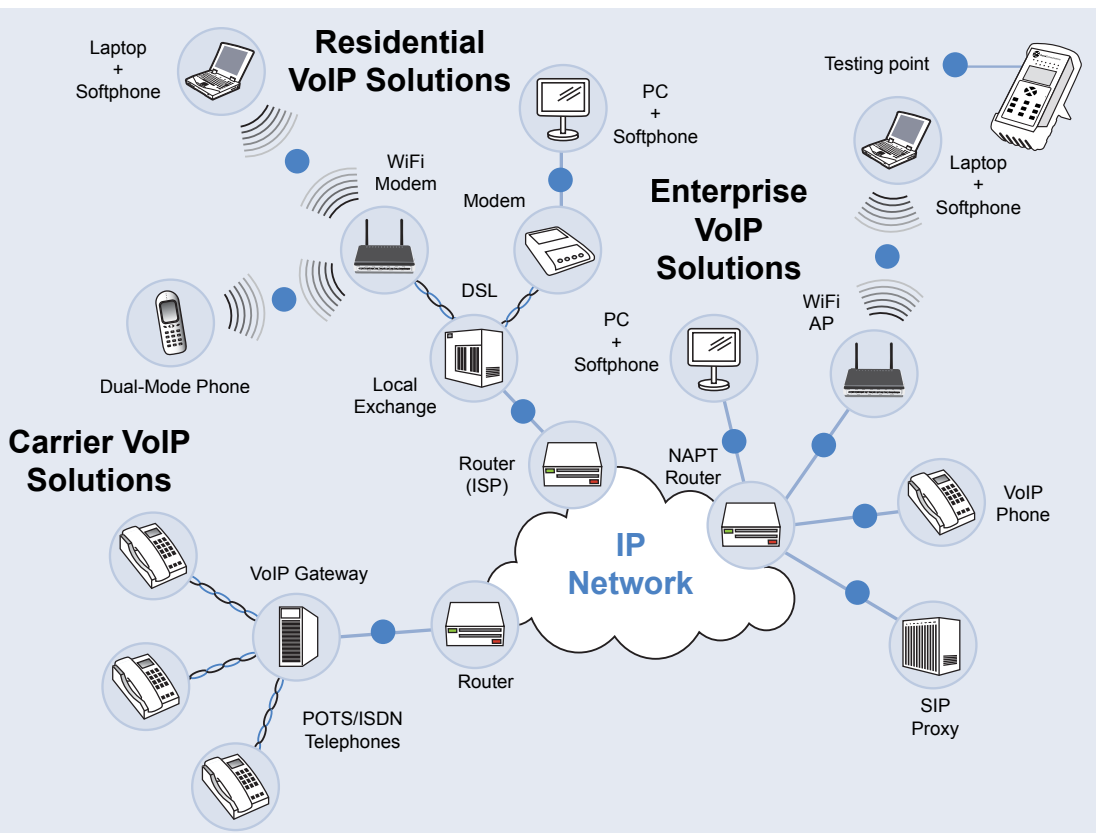
# TrendUnipro VoIP

Compact Testing Solution for VoIP and VoWLAN



The required revenue from new Carrier Class VoIP services and cost savings expected from Enterprise VoIP will not be realised if low network performance degrades the customer perception of the service. The isolation and diagnosis of potential VoIP service faults is therefore essential for user satisfaction and the success of the service.

IP Telephony service verification is now possible in a few seconds, thanks to Trend Unipro VoIP. This tester is a compact, autonomous and easy-to-use tool that checks the quality of the voice, and rates the service by estimating the Mean Opinion Score (MOS) with the help of the popular ITU-T R-Factor metric. Measurements can be carried out either over wired or wireless connections. Ethernet and WiFi interface options allow access technology-independent testing of the delivered service.



- Emulation of wired and wireless VoIP devices
- R-Factor and MOS
- RTP jitter, loss and delay indicators
- Quick boot-up time
- SIP message trace
- STUN support for NAPT pass-through testing
- Data tests including FTP and HTTP service verification
- End-to-end connectivity tests including Ping and Traceroute

# Test and Measurement

For Quick and Easy VoIP Service Rollout

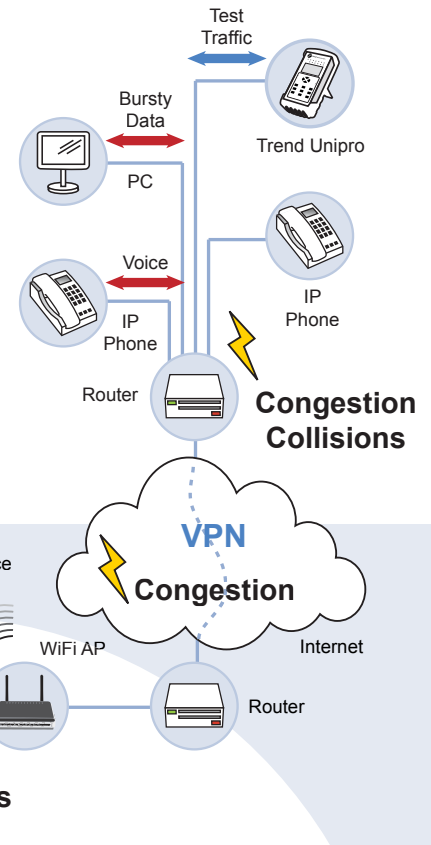


## MOS Rating and QoS Testing

VoIP or IP telephony is both a low-cost alternative to traditional Analog and ISDN telephony and an application that can utilise existing IP data infrastructure to deliver voice service. However, maintaining the same quality and reliability that became routine with digital circuit-switched telephony is a formidable challenge.

The MOS and R-Factor are widely recognised QoE metrics for VoIP, used to rate the service end-to-end. They take into account encoding/decoding/transcoding processes, network congestion effects, interferences in wireless channels, and any other possible source of degradations. To exercise the service, an IP Phone emulation is provided which enables subjective service quality evaluation, complementary to the objective QoS metrics.

Trend Unipro VoIP also evaluates QoS based on RTP statistics during calls to and from the tester, in addition to the optional advanced MOS and R-Factor metrics. These metrics enable the call to be benchmarked against predefined thresholds to give a simplified Pass/Fail result.



QoS	QoE
Delay	MOS
Jitter	R-Factor
Packet Loss	



## A Versatile and Manageable Tester

Trend Unipro VoIP is a rugged, battery-operated tester that may be powered up and ready for use in less than 15 seconds. These features make installation tasks faster and easier in field operations or in any other testing environment.

The easy-to-use Trend Unipro user interface reduces the training needed to use it to the minimum. User-definable profiles speed up and enable easy configuration of the unit for testing in different locations. Supervisors and power users can use the innovative web-based interface for remote configuration, testing, and extensive reporting.

Using the Trend Unipro VoIP it is possible to quickly isolate faults and determine what is failing in the network. Put simply: with the help of Trend Unipro VoIP your customers can quickly forget that you are using the Internet Protocol to deliver their high-quality telephony services



# Get the Maximum

Out of your Converged Voice & Data Network



- Remote control with web-based interface and DHCP server
- Connectors for external headset and microphone
- User profiles for the WAN interface, SIP service, call tests and data tests
- Long battery life
- Result and configuration sharing with USB memory device
- IP, Ethernet and WiFi statistics

## WiFi Interface for VoWLAN Testing

Many network devices in use by businesses and home users are increasingly being connected using WiFi. Also, new dual-mode mobile handsets use VoIP and wireless access points to extend coverage of traditional cellular phones.

The Trend Unipro VoIP WiFi interface supports the IEEE 802.11b/g wireless standards to deal with the specific issues that are involved with transmission through the air interface. This feature becomes a powerful one when it is combined with the internal battery operation. When used in the field, Trend Unipro VoIP's long battery life and WiFi interface offer unparalleled VoIP testing flexibility.

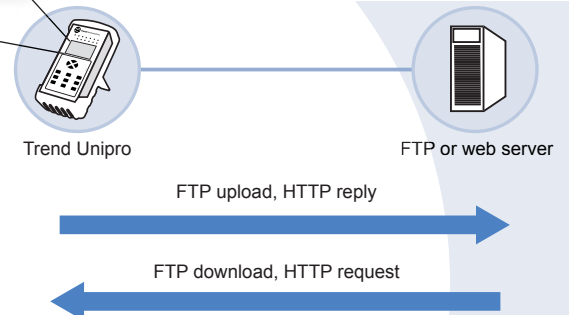
## HTTP and FTP Service Verification

VoIP brings together data and telephony services. When both kinds of applications are run over the same network, it is necessary to check that voice services are not disturbed by data traffic and vice-versa. While testing SIP, Trend Unipro VoIP can test whether web and FTP services can run without impeding the quality of the VoIP service.

The HTTP and FTP test capabilities complete the common test requirement for converged voice & data networks and make Trend Unipro VoIP a uniquely valuable tool.

```

HTTP Download
Status Disabled
Dest. www.trend...
Filename index.html
Rx Bytes 106
Rx Time 12
Rx Rate 33
    
```



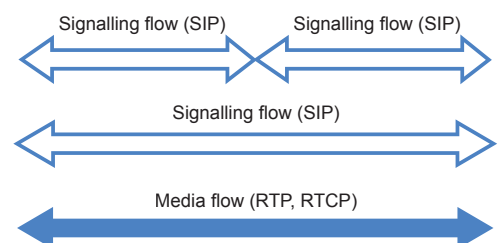
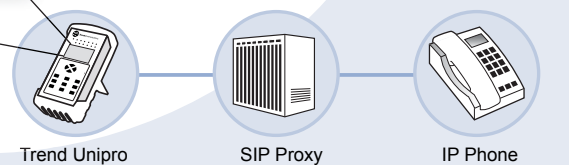
## Analysis of Signalling-Related Issues

Trend Unipro VoIP captures and displays the SIP signalling messages associated with the VoIP test being performed. Trend Unipro VoIP enables quick capture and analysis of the SIP protocol to help with troubleshooting common signalling and configuration problems.

Trend Unipro VoIP includes STUN protocol support which simplifies the troubleshooting of NAT router signalling issues that can block calls.

```

SIP Trace History
Tx 0.000
REGISTER sip.pbxtrendcomms.com
Tx 5.267
INVITE sip4545@pbxtrendcomms.com
Tx 79.252
REGISTER sip.pbxtrendcomms.com
    
```



## TrendUnipro VoIP

<b>Test Interfaces</b>	10/100BASE-T Ethernet full duplex IEEE 802.11 Wi-Fi
<b>VoIP Tests</b>	VoIP phone emulation Multicall test SIP message trace QoS test (packet loss, delay and jitter) MOS and R-Factor analysis
<b>Data Tests</b>	Ping and Traceroute FTP upload and download HTTP download IP, Ethernet and Wi-Fi statistics
<b>Protocol Stack</b>	Support of Ethernet VLANs as per IEEE 802.1Q Support of WEP and WPA protocols for privacy over wireless interfaces DHCP client for automatic configuration of the IP layer PPP over Ethernet with CHAP and PAP authentication VoIP signalling protocol: SIP Voice codecs: ITU-T G.711 (A-law and $\mu$ -law), G.729, G.726, G.723
<b>Functions</b>	Support of STUN and outbound proxy for NAPT transversal Auto-answer/unattended mode for VoIP calls MOS and jitter vs. time graphics Configurable user profiles for the WAN interface, SIP service, call tests, and data tests
<b>User Interface</b>	OLED-based, 6 line keypad-driven local user interface Web-based remote user interface
<b>Storage</b>	Internal storage for 50 results Result export to USB or FTP
<b>General</b>	Ethernet PC port for remote control and data import/export External and integrated microphone and speaker Field upgradable from PC, USB disk or FTP server Result storage and export to USB disk or FTP server
<b>Ergonomics</b>	Size: 90 mm x 140 mm x 45 mm Weight: 700 g (1.5 lbs) Battery life: 8 hours Boot-up time: < 15 s

